AN AUDIO DISPLAY
OF
PROCESSED ACOUSTIC SIGNALS

George Liell Mager

NAVAL POSTGRADUATE SCHOOL
Monterey, California

THESIS

AN AUDIO DISPLAY
OF
PROCESSED ACOUSTIC SIGNALS

by

George Liell Mager, Jr.

June 1974

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An Audio Display
of
Processed Acoustic Signals

by

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Submitted in partial fulfillment of the
requirements for the degree of

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ABSTRACT

An audio display system using eight tones to encode the output of a digital signal processor is investigated. The purpose of the system is to detect and display an audio tone masked by bandlimited Gaussian noise. The digital processor performs a spectral analysis on the signal using a Fast Fourier Transform. Several hundred experiments were performed using a human listener to monitor the display. The experimental system is capable of displaying the difference between signal, no signal, and signal plus noise conditions. About 80 percent probability of correct decision by the human observer was achieved for input signals below the detection threshold for an unaided listener. Possible applications to a sonar system are indicated.

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I. INTRODUCTION

A. AUDIO DISPLAYS AND THEIR USE WITH SONAR

In almost every man-machine system there must be some sort of interface or display in order that the operator may know what is going on in the system. The display is a means of providing information the operator can not or does not receive directly through his senses.

Sonar systems require such a display to interface with the operator. Sonar utilizes sound energy as a means of locating submerged objects in the ocean. Active sonar transmits the sound energy into the water and then depends upon the returning echoes reflected off the target of interest for bearing, range, and other information. Because of its acoustic nature, it seems natural that part of the sonar display system should include audio information. Older sonars, utilizing frequencies in the high sonic and ultrasonic ranges, presented an audio display of the echo return hetrodyned down to a frequency suitable for hearing by a human operator (about 800 Hz). This audio display provided valuable information which could be used in target detection and classification. Newer sonars, using lower transmitting frequencies, have lost some of their ability to provide audio information which is of any use in an unprocessed form. For example, one displayed quantity of great use in sonar is the shift in the frequency of an echo due to the Doppler phenomenon. This is the result of the target moving relative to the background of its environment. Older sonars

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operating in the vicinity of about 20 kHz displayed a Doppler shift of about 40 Hz for a three knot target. This frequency shift would be maintained when the signal was hetrodyned to 800 Hz, and could be easily heard by the human operator. In order for a newer sonar, operating at three kHz, to exhibit the same Doppler shift, the target would be required to have a speed of about 20 knots. A three knot target would exhibit a shift of only about six Hz, which would be much more difficult to hear, even when hetrodyned to 800 Hz. [1]

As sonars have increased in complexity and greater use has been made of advanced forms of signal processing, the use of the audio information has declined somewhat and an increased emphasis has been placed on the development and use of visual presentations. Even with older and somewhat less sophisticated sonars, operators have a tendency to rely upon the video display for detection, using the audio information only as a means of target classification.

A video display provides the operator with valuable information, primarily due to a longer integration time and longer memory than is

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[1] Calculations based on the simplified equation [1, 2]

$$f = \frac{2f_o u}{c}$$

where,   f   is the Doppler shift
           f_o is the transmitted frequency
           u   is the target velocity
           c   is the velocity of sound in sea water, approximately 1500 m/sec.

usually possible with the audio display. It is invaluable in target tracking; the PPI presentation provides an easy means of displaying range and bearing. But a CRT does not provide all the information necessary to evaluate target aspect, size and classification. Detection is difficult, at best, on a video display. Camp [2, p 271] states;

> "Displays have been a weak link in the sonar detection process since the first attempts to make video displays. The insufficient dynamic range, the failure to provide an optically resolvable location for each processor resolution cell, and the failure to provide equal areas for all resolution cells have degraded sonar performance several decibels relative to what is possible ... The plan position indicator (PPI) is a good example of a display which is susceptible to all the inadequacies listed above. This fact does not mean that plan position indicators should not be employed in sonars, but they probably should not be used in the detection process."

A further problem associated with the video display is the requirement for constant attention. This makes it undesirable in situations where an operator is unable to give the display his undivided attention. The display can be fatiguing when watched for hours on a long and inactive patrol, especially in the darkened room required for proper operation of a CRT display.

B. DETECTION OF A LONG RANGE SONAR TARGET

If there were nothing more involved in the use of sonar echo ranging equipment than hearing a reflected pulse, sonar systems could be quite simple. Unfortunately, this is not the case, and many factors such as spreading and attenuation of the acoustic energy, noise of all

sorts, and masking by reverberation combine to degrade the echo and make it more difficult to hear. Reverberation, a major source of difficulty in receiving a target echo return, is caused by energy returned by reflectors other than the target. These reflectors include all scatterers within the volume of the ocean such as fish, bubbles, suspended matter, etc. The ocean's surface and bottom also cause reverberation. When the reverberation energy is of comparable magnitude to that returned by the target of interest, the echo will be masked and the sonar is said to be reverberation limited.

Increasing the power output or source level of the sonar is of no use in attempting to overcome a condition of reverberation limiting since both the echo level and the reverberation level increase directly with the source level [1, 3, 4]. On the other hand, although the reverberation and target levels are both decreased by attenuation with increasing range, the target echo level is attenuated at a faster rate than the reverberation level [1, 3, 4]. Thus, some method other than a source level increase must be used to detect a target buried in reverberation.

C. PURPOSE OF RESEARCH

The purpose of the research reported in this paper was to investigate a possible method of detecting a target at long range whose echo is masked by reverberation, and to develop a method of alerting the operator to the presence of the possible target, using an audio display. Such a system would have several advantages. First of all, if a "black

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box" method of detecting a target masked by reverberation at long ranges could be developed, it could be added to existing sonar systems to improve their performance.

Secondly, an acoustic display of the correct form might be used to alert the sonar operator of the presence of a possible long range target even when he was concentrating his search effort in closer and higher threat areas. The display might provide information allowing correlation with video or conventional audio displays. If properly designed, the display might be usable in areas such as the bridge or CIC. An audio display could provide another information input to be used by the operator in his decision making process without adding another video channel.

II. CONCEPT OF THE RESEARCH

A. CHARACTERISTICS OF REVERBERATION AND DOPPLER SHIFT

Reverberation does not usually lie at the same frequency as the transmitted pulse but is shifted in frequency due primarily to the velocity of the sonar platform. In addition, the reverberation is spread in frequency due to the finite duration of the sonar pulse (1/t Hz spread, t = pulse length), the variations in Doppler due to reverberation echoes arriving from different directions, and the motion of the scatterers [1, p 228]. This frequency spread is less for lower frequencies [5, p 330]. Because of the frequency spreading effects, a spectrum of band limited reverberations might appear as in Figure 1. This would be centered somewhat around the transmitting frequency but shifted due to the Doppler effect of own ship's motion.

If a target were present, its spectrum might appear as in Figure 2 depending on target aspect. If there were no Doppler shift, the target and the reverberation would occur at the same center frequency and the target could be undetectable unless some increase in intensity occurred. However, assuming that there were some motion of the target relative to the background, there would be some Doppler shift. If, for example, the Doppler effect caused the target echo to shift upwards in frequency ("up Doppler"), the target would appear as an increase in the magnitude
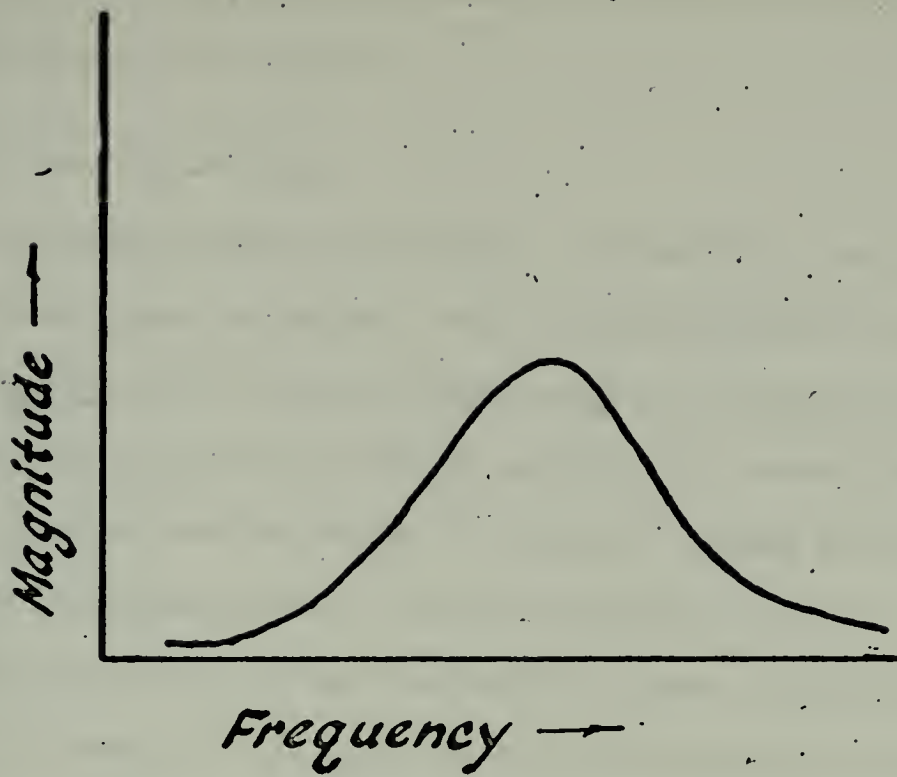
Figure 1. Bandlimited Reverberation Spectrum
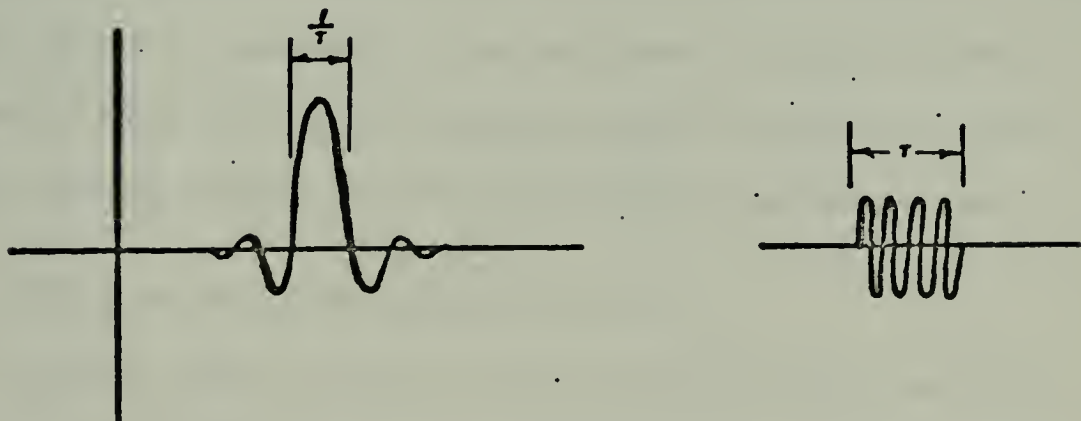
Figure 2. Target Echo Spectrum Showing Relationship Between
(a) Spectrum and (b) Pulse Length

of the spectrum at a frequency away from the center of the spectrum of reverberation as in Figure 3.

B. DESIGN OF DISPLAY

Recognizing these characteristics, it is possible to envision a system which would make use of them. Such a system would take a sample of a sonar return during a specified time segment of the ping cycle corresponding to a range band of interest, and form its frequency spectrum. This spectrum would be divided into segments, making use of a sliding window to pick out a portion of the spectrum of interest. For example, 12 segments of the total spectrum might be chosen as illustrated in Figure 4. Each chosen segment would then be encoded by using its magnitude (C_n) to control the amplitude of an individual oscillator. The outputs of the individual oscillators would be summed, and the resultant summation would become the display. The display, if properly encoded would spread the spectrum as shown in Figure 5. Also, as shown, a threshold could be set so that magnitudes below a certain level would not be encoded, thereby reducing interfering and confusing sounds.

C. ADVANTAGES OF PROPOSED DISPLAY

If such a system were used, several effects could be predicted. First of all, since the oscillators may be encoded in any manner, problems with audio frequency discrimination might be avoided. It would only be necessary to ensure that each frequency segment was
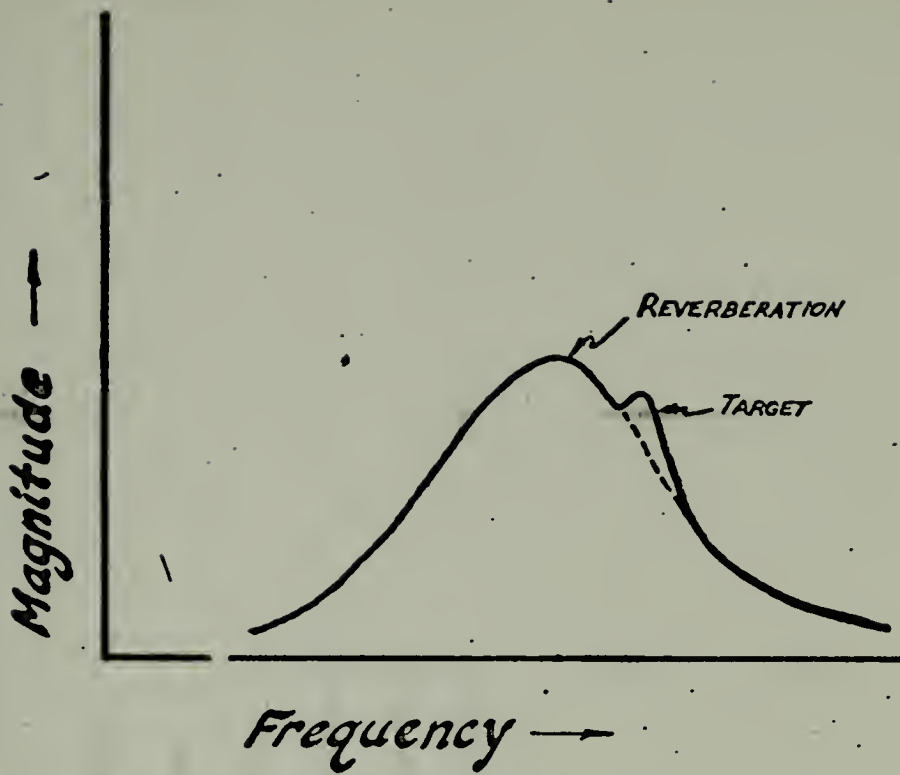
Figure 3.   Up Doppler Target and Reverberation
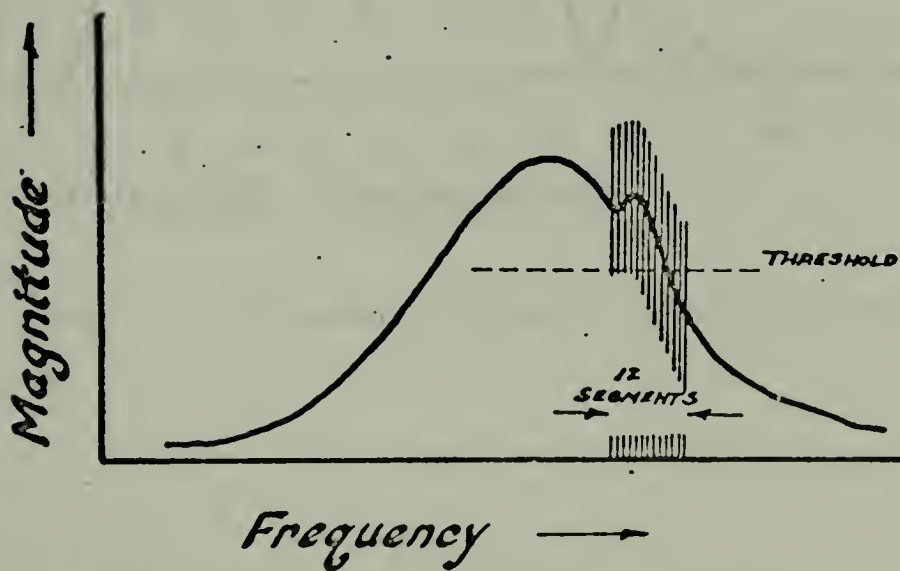
Figure 4.   Segmented Frequency Spectrum

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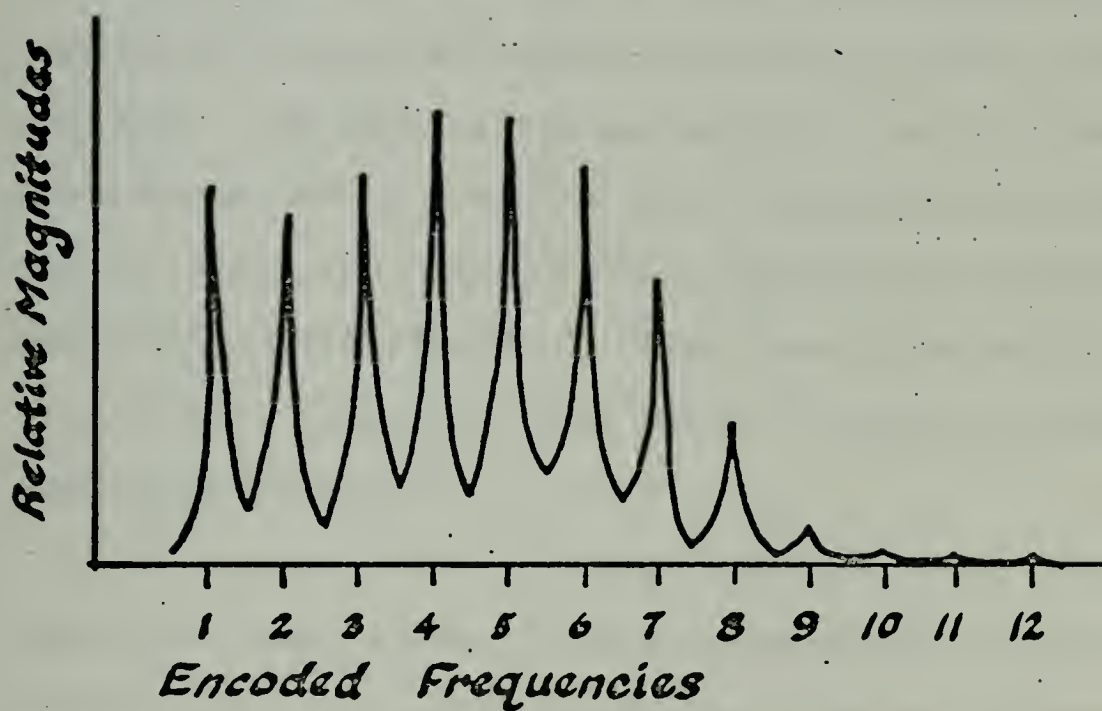
Figure 5.   Expansion of Frequency Spectrum by Encoding

sufficiently removed from adjacent tones to avoid confusion. The fre-
quencies chosen for encoding need bear no direct relation to the fre-
quencies in the original signal. Chapnis, et al [6], suggest that an audio
display should show some relationship with the information it is display-
ing, i.e., it should sound like what is happening. Therefore, it might
be advantageous to encode in such a manner as to retain relative posi-
tions on the frequency scale. In this way, an "up Doppler" target whose
frequency was higher than that of the reverberation would logically be
encoded with a tone higher in frequency than that used for the reverbera-
tion center. The human ear has been modeled as a series of relatively
high Q bandpass filters, each with a certain critical bandpass character-
istic [1]. The critical bandpass describes the maximum bandwidth of
noise which will mask a pure tone. Urick and others note this to be
about 50 Hz at a frequency of 500 Hz, and that it increases in bandwidth
with increasing frequency.

A raw audio data display containing a contact with only a small
Doppler shift would be likely to have its entire useful spectrum fall
completely within one bandpass resolution cell of the ear and therefore
the target would be completely masked by the noise. Even if the target
were not masked by the noise, the Doppler shift might be less than the
pitch discrimination ability and pitch memory of the operator.

If, however, the display system was encoded so as to increase the
frequency difference between the target echo and the reverberation

center, the problem of critical bandwidth could be avoided. The target would be encoded with an audio tone different from that of the reverberation center. Thus, the resolution cell size is effectively reduced and the Doppler shift is enhanced.

A third advantage of the proposed system is that it would allow time expansion. One of the problems encountered in sonar is that, in addition to the fact that a target echo is usually extremely short in duration, possibly too short for true audition, the period between the echoes is exceedingly long if the target is at long range. If the velocity of sound in sea water is taken to be 1500 m/sec., there is a period of almost 27 seconds between successive detections of a target at 20,000 meters. Thus the human operator searching for a target at long range is placed in the untenable and unenviable position of trying to detect through a heavy noise background a sound of approximately the same frequency as the background. The target echo duration may be so short as to make it nearly impossible to detect and its repetition rate is so slow as to make it difficult to remember that an echo was in fact heard. The brain is unable to remember the echo long enough to compare successive detections.

III. EXPERIMENTAL PROCEDURES

A. DESCRIPTION OF EXPERIMENTS TO BE PERFORMED

The experimental phase of the research consisted of modeling the display system using a hybrid computer system incorporating the CI 5000 Analog and XDS 9300 Digital computers installed at the Naval Postgraduate School in order to learn what such a display would sound like. It was not known beforehand if the system would provide a useful information display. The task, therefore, was to design a hybrid computer program which would accept an input analog signal, convert it to a digital format, perform a Fast Fourier Transform (FFT) spectrum analysis, and then use the resulting spectral information (magnitude coefficients) to control the amplitude of a number of discrete oscillators on the analog computer. The output of these oscillators was to be summed and then presented to the operator via headphones or a loudspeaker.

With this phase completed, the next segment of the experiment would be to test the system using various inputs, such as broadband noise, pure sinusoids, pulses, and various combinations of these to learn

     a. Whether the system would work as expected, and

     b. Whether any useful information could be learned from the display.

Since the system as herein envisioned is relatively independent of the actual characteristics of the sonar pulse, the audio display may be held as long as desired either by providing for operator control or by using automatic cycling. For example, the display might be set to last the length of time between successive detections or successive looks at a given range. This would allow comparison of successive displays. Or, it might be possible to examine several different frequency windows at a given range, during the period between detections.

Questions to be answered included:

a. Is the display feasible? i.e., does it present any information at all, or does it merely sound like more noise?

b. Is it possible to detect a signal in noise with the system, and if so, is the signal-to-noise ratio of the system better than the raw data?

c. Does the encoding facilitate detection, and what is the best encoding scheme?

B. DESCRIPTION OF COMPUTER PROGRAMS

1. Analog Program (Display)

The first part of the system modeling phase was the building of the oscillators and other associated analog control circuits. Because of patchboard limitations it was possible to construct only eight oscillators. Each consisted of three analog amplifiers connected as shown in Figure 6.
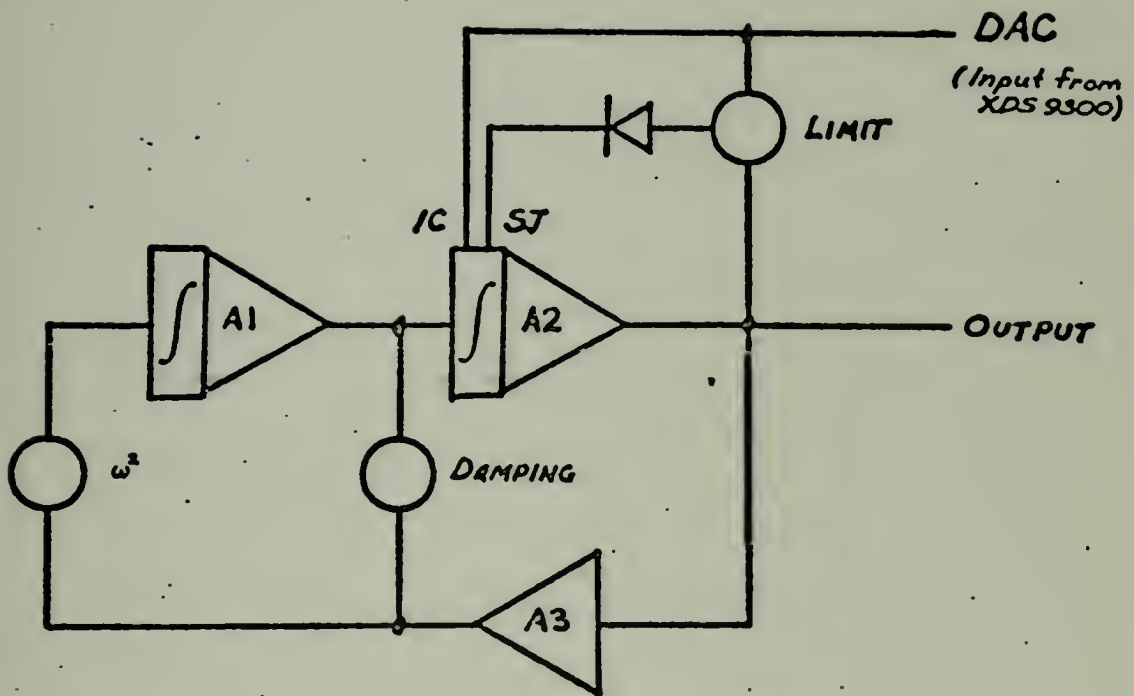
Figure 6.   Diagram of a Typical Analog Oscillator

This circuit, with amplifiers A1 and A2 connected as integrators and A3 as a feedback amplifier, is basically a cosinusoidal generator with potentiometer P1 controlling frequency, (^{2})., P2 controlling damping, and P3 limiting the output. The diode aids in limiting the positive feedback and also allows better control of the amplitude from the DAC line. Because none were available on the computer patchboard, these diodes were added as an external component.

The DAC input is the digital-to-analog level information from the digital computer, and, in this case, represents the magnitude of the spectral components as computed by the XDS 9300.

The outputs of the function generators or oscillators were then summed and the output of the summing amplifier was fed through a potentiometer, used to control the overall level of the display. External to the computer, the signal was fed through an audio amplifier and thence to a speaker or headphones. See Figure 7.

2. Digital Program (Signal Processor)

The basis for the digital program used in the system was a computer library subroutine called FOUR 2. This program performs a Fast Fourier Transform or FFT. Reference 7 contains an excellent description of the theory behind the FFT and spectral analysis as performed on a small computer of the type used in these experiments.

FOUR 2 requires digital information in order to perform the FFT. A program was available on cards at the computer center which
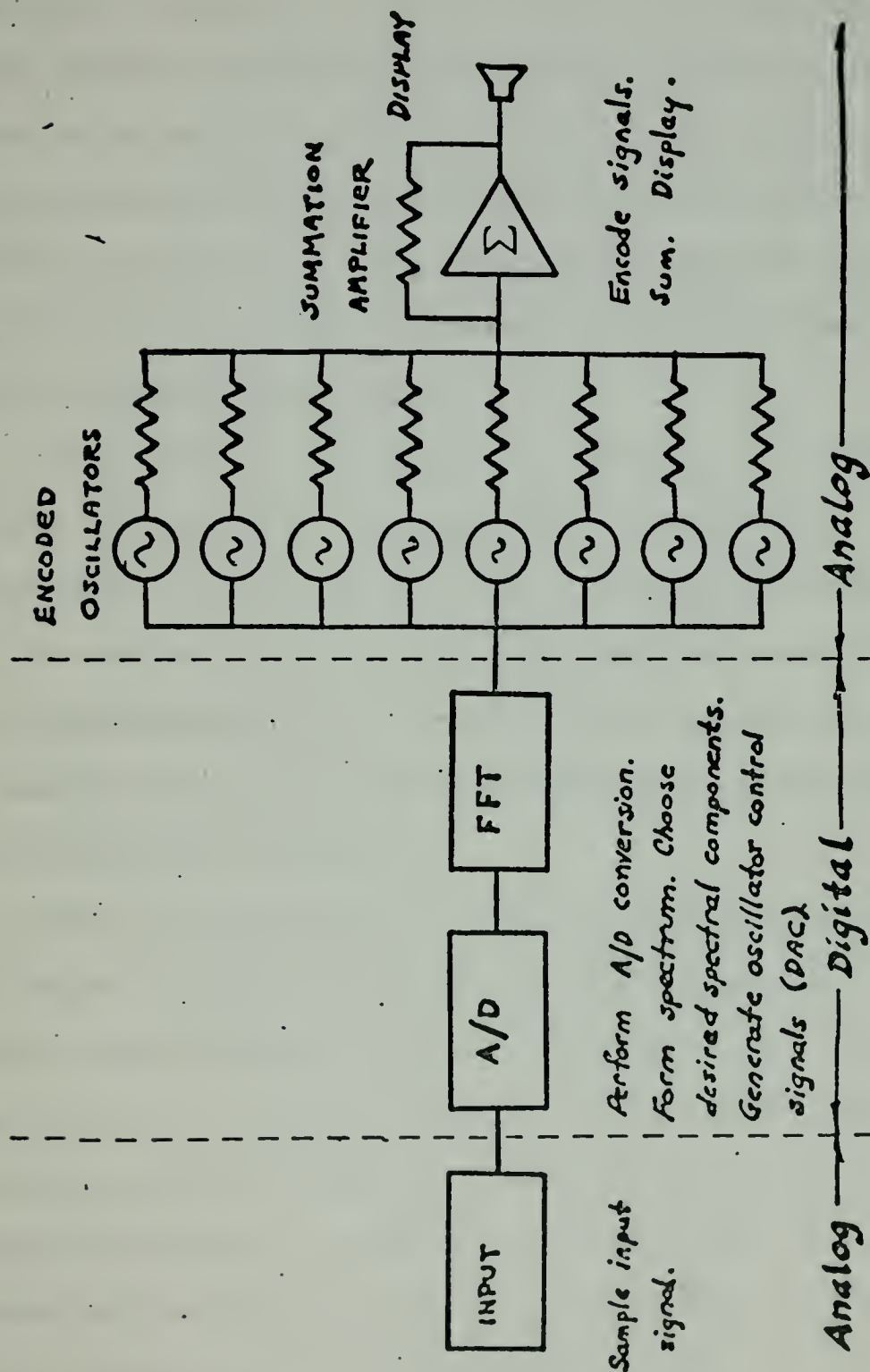
Figure 7. General Form of the System

would perform the Analog to Digital conversions. The program, called simply A/D, is somewhat more involved than is necessary for this application in that it is capable of performing more than just the A/D conversion. A number of other options are available which, while not used, were not removed from the program since they might be of later use. Several of the options, for example, make use of the fact that the program records the input data on magnetic tape, which would allow a given sample input to be examined more than once.

To this A/D conversion program, a subroutine called PROCESS, was added. This subroutine, as initially written, called FOUR 2 to perform the FFT on 1024 samples of the input. This limit was set by the size of the computer memory. The spectrum was then formed by computing the square root of the sum of the squares of the computed components. The resulting total spectrum contained 512 spectral components, by virtue of the method of computation.

Since there were only eight oscillators available, only eight of the spectral components could be chosen. While it might have been preferable to have this choice made by external input so that different components could be chosen for examination at will, this was not done. Referring to the previous conjecture that if a signal were present, the magnitude of its frequency component would be greater than those of the immediately surrounding components, it was decided to have the program compare all of the component magnitudes and choose the

greatest. This component was called the center frequency (SPECMAX in the program). Based upon this decision, the computer then chose the three adjacent lower components and four adjacent higher components, and the eight components thus chosen were used to set the DAC inputs to the analog computer. Since the DAC inputs are required to be between 0 and 1, the maximum component was divided into all eight components in order to normalize them. In this manner, the center frequency would always have a magnitude of 1 and the others proportionately less.

The subroutine was written to allow the operator to control operation of the digital program at the analog computer console. Three toggle switches on the console were tagged TEST 1, TEST 2, and TEST 3. When TEST 1 is activated the program is initiated. The A/D conversion of the input analog signal is made, the FFT is performed and the DAC lines are set, which presents the audio display output. The output is held until TEST 2 is activated, at which time the program returns to the main program at the point where the process may again be activated by TEST 1. TEST 3 allows return to the main program at the point where the program options are entered at the TTY. This function was provided for future expansion but was seldom used.

Additionally, the capability of setting the frequency and damping pots was written into the A/D program, and, in order to

monitor the program results, the DAC line settings were made available at the line printer.

The program, and instructions for its use are contained in Appendix C.

C. INITIAL TESTS AND EXPERIMENTS PERFORMED USING ANALOG PROGRAM

For initial testing of the oscillators, the potentiometers controlling limiting and damping were manually controlled while the frequency potentiometers were servo controlled, being set to previously calculated values. When the analog program was later used with the digital computer controlling the operation, the damping pots were also servo controlled, and, like the frequency pots, were set by the digital computer to previously determined values.

For initial testing, prior to the inclusion of the digital program, the DAC inputs were set using an on-board 100 volt line and manual potentiometers.

The first test performed consisted of setting the oscillators to frequencies from 300 to 1000 Hz, in 100 Hz steps. This particular frequency range was chosen for two reasons. First of all, it has been shown that the most sensitive region of frequencies for human hearing is in the vicinity of 500 to 1000 Hz. [1, 4]. Secondly, the CI 5000 is limited in frequency by the values of the integrator multiplier capacitors available [8]. In the case of the circuit used, the

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maximum frequency which may be generated is in the vicinity of 1500 Hz.

Initial limit potentiometer settings were somewhat arbitrary, and the damping potentiometers were set to provide stable operation of the oscillators. Refer to Appendix A for settings used.

It was first noted that when all eight oscillators were set to have a zero input on the DAC lines, it was possible to hear a mixture of tones. These resulted from internal noise in the computer, which caused the oscillators to oscillate at a low level.

When the input to any particular oscillator was increased, the resulting tone from that oscillator quickly masked the noise from the other oscillators. When any two tones were set to approximately equal levels, additional beat notes at other frequencies were formed, as would be expected.

As the number of frequencies was increased, it became increasingly difficult to discern individual changes. When seven tones of roughtly equal amplitude were present, it was virtually impossible to tell when an eighth tone was added until its amplitude was significantly greater than the other seven.

This masking effect has two origins, both of which are described in texts on experimental psychology. The first cause is that of masking of tones by other tones, similar to the problem encountered by the sonar operator attempting to hear a sonar echo masked by reverberation. Thus when seven tones of approximately equal amplitude

are present, their combined effect is to raise the threshold level necessary in order to hear the eighth tone. On the other hand, when the eighth tone is of sufficient level, the reverse effect occurs, i. e., it has a masking effect on the other seven tones.

A second effect has been described as a phenomenon of pitch perception [9]. If a subject is presented with a combination of tones which are harmonically related, as they were in this case, a phenomenon known as the "missing fundamental" occurs. Evidently the ear generates tones which correspond to the lowest common denomenator, or fundamental, of all the tones present in the complex sound, and adds that tone to the original combination of tones. Thus, even though there are actually only seven tones displayed, the ear may hear other tones which are not actually present. These also add to the masking effect.

In an attempt to obtain data of a more quantitative nature, all levels were set to provide a 40 volt peak-to-peak indication on the installed oscilloscope, with a DAC input of 100 volts, and damping pots were again set to provide stable oscillations. Frequency settings remained the same. It was noted, however, that actual frequencies varied as much as five percent from calculated values.

DAC lines to each oscillator were again routed through potentiometers to simulate the input from the digital computer. The DAC line to oscillator number five was then set to 100 volts (maximum) with

all others set to zero. The resultant output was a relatively pure sine wave.

When a signal was added on lines four or six, it was possible to recognize a change in the tonal pattern when the level of the added signal was only one-tenth that of the reference tone. When two tones were added, it was still possible to tell that a change had occurred. However, as more tones were added, it became increasingly difficult to recognize any change in the pattern.

Since these experiments were of an inherently psychophysical nature, it was recognized that the person performing the experiment probably had a mental bias since he knew what he was looking for. Therefore, at this point in the experiment, an observer with no knowledge of the system was asked to listen while changes were made.

With all DAC lines initially energized, each line input was reduced to zero until all oscillators except number five were deactivated. Not until this point was reached did the observer indicate that he noticed any change, realizing at that time that only a pure tone was present.

On the second trial, the process was reversed. In this case tones were added one at a time and the subject was asked to comment on what he heard. He was aware that tones were being added. However, when oscillator number five was on (frequency 700Hz), the addition of a tone from oscillator number eight (frequency 1000Hz) elicited

a comment from him that a low frequency tone was being added. The experimenter repeated this same experiment on himself and had no difficulty in determining that a higher frequency tone was being added to that of oscillator number five. The disappointing result of this experiment is difficult to explain. Certainly, the observer was not faced with the mental bias of the experimenter and therefore reported what he thought he heard. But this does not explain the fact that he reported hearing a tone lower than the reference tone when in fact, a higher tone was being presented. Possibly a lack of understanding of the object of the experiment is the answer. In addition, it is assumed that if such a display were actually put to use, operators would be trained to listen to certain changes.

In order to remove the harmonic relationships previously noted as having been the possible cause of some of the problem, the oscillators were tuned to a musical scale. In addition to the other problems noted, the use of a 100 Hz spacing resulted in very discordant combinations which were very difficult and tiring to listen to. Appendix B lists the musical scale and associated frequencies as listed in Reference 10. Also included are the calculated ^{2} and potentiometer settings. Using the notation of Reference 10, the whole notes of the octave from G to G^1 were encoded, which centered the encoded tones around 500 Hz, with an average spacing of about 55 Hz. On this scale, the spacing between whole notes is in a ratio of either

When all eight oscillators were added together, the resultant combinational tone was still somewhat discordant, but was easier to listen to than when 100 Hz spacing was used. It sounded similar to that which would result from playing the eight chosen notes simultaneously on an electronic organ.

Oscillator five was again set at maximum and used as a reference tone while the other tones were added. The immediate result noted was that, as would be expected, certain combinations produced musical chords, and therefore relatively distinct patterns. Other combinations did not produce these chords and therefore it could be said that different and distinct patterns were being produced.

Amplitude variations of individual tones were not always recognizable unless they were relatively large. This would tend to agree with published results [1, 4, 6, 11, 12]. It was also noted that variations in both amplitude and frequency pattern were more obvious when headphones were used instead of the loudspeaker.

In attempting to recognize distinct tones, those which were widely separated were easier to discern than those which were closer together, again agreeing with published data [1, 4, 6, 11, 12].

D. COMBINED DIGITAL AND ANALOG EXPERIMENTS

Upon completion of the experiments conducted on the Analog computer alone, the digital program was added and the remainder of the experimentation was performed on the entire system.

1. Initial System Tests

The first series of experiments used an input of a pure sinu-soid with a frequency of 3 kHz. In each case the result was roughly what was expected. The audio display, in every repetition of the experiment, produced a strong central frequency. Occasionally there would be weaker background tones, and the line printer output indicated that other tones were being produced. Generally, the strongest of these were those on either side of the center tone. Their magnitudes were never greater than about one-third that of the center tone.

The analog board was wired in such a manner as to allow monitoring both the composite output and the outputs of the individual oscillators on both oscilloscope and via the audio amplifier output. Comparison of the measured outputs of the individual oscillators with their DAC inputs as recorded on the line printer confirmed that the oscillator amplitudes were being correctly controlled by the DAC lines.

It was noted that different frequency inputs produced different results. In addition, the results obtained were not always identical. For example, on some occasions a pure sinusoidal input would pro-duce a single tone output while on other occasions two or three tones might be produced. This is believed to be a result of

1. the sampling rate (10 kHz) which determines the resolution of the FFT.

2. system noise, and

3. system instabilities.

In the next experiment, broadband noise was added, using the noise generator available at the analog computer. This caused a definite change in the display. Because the noise was random (Gaussian), its spectral components were not constant, and, as a result, each display was composed of different combinations of the eight tones. However, the central tone was still dominant and was easily discernable.

Finally, noise alone was sampled. The result was similar to the discordant set of tones described earlier. Some tones would dominate each display, but, unlike the case where noise and the sinusoid were both present at the input, the pattern which was produced was not constant, and had different combinations of tones present for each display.

Table I illustrates three typical repetitions of the experiment. Using the data of Table I the graph shown in Figure 8 was constructed. No measurements of relative amounts of signal and noise were made at this stage of the experimentation. Realizing that it may not be realistic to consider a graph based on only eight data points as being reliable, it is still possible to observe the relative differences between those cases where the sinusoid was present and where it was not. When noise was added to the sinusoid, the shape of the curve

	1	2	3	4	5	6	7	7
Sinusoid Alone	0.1315	0.2094	0.5280	1.0000	0.2555	0.1460	0.1002	0.0757
Sinusoid Plus Noise	0.1435	0.2547	0.6657	1.0000	0.2859	0.1510	0.1098	0.0976
Noise Alone	0.2770	0.5856	0.2728	1.0000	0.3151	0.0455	0.1808	0.2704

TABLE I

Typical DAC Line Settings for Various Input Conditions

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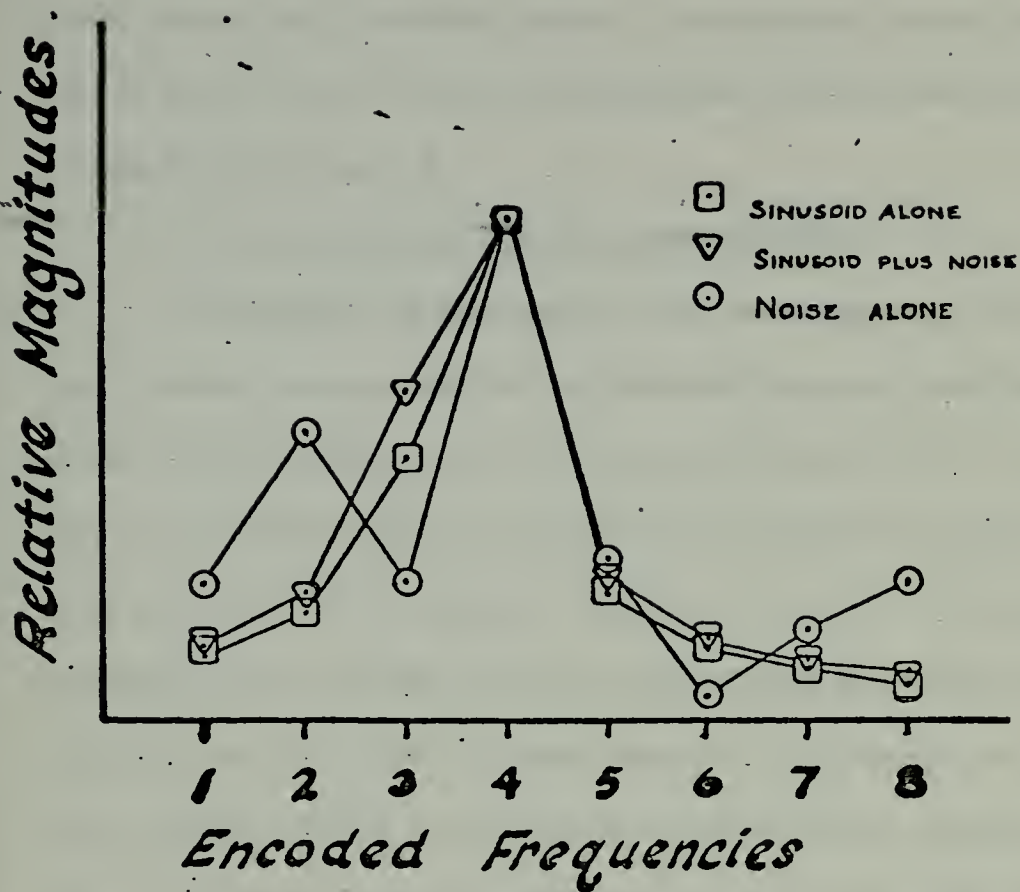
Figure. 8.   Plot of DAC Inputs to Encoded Oscillators
for Three Different Input Cases

was essentially unchanged from the pure sinusoid case, the curve peaking at the center tone and falling off rapidly to either side. However, when only noise was present, the curve was irregular, with peaks occurring at random places. This graph is typical of the results which were obtained throughout the experimentation and should be compared with Figure 5.

2. Tests on System with Expanded Encoding

The oscillator frequencies were next changed. In order to gain a wider frequency difference between the tones, and thus aid in auditory discrimination, every other tone from C to C^2, a spread of two octaves with no harmonic duplications except at the ends, [10] was used to encode the oscillators. Since 1024 samples are taken at a sampling rate of 10 kHz, the resolution of each sampling cell is slightly less than 10 Hz. If eight samples are encoded, the total system sampling window is approximately 80 Hz wide. By using eight notes, spread over the indicated two octaves, the displayed bandwidth becomes approximately 785 Hz. Thus there is roughtly a 10 to 1 frequency expansion, with an average of 98 Hz per resolution cell.

Following these changes, a second series of tests was performed on the system. Each test was performed three times and the results were as noted below. Note that a second signal generator was added as a signal source.

Test 1. Input: A single 3 kHz sinusoid.

Results: All of the displays were similar, sounding like a combination of several tones.

Remarks: With no reference with which to compare the display, it was sometimes difficult to pick out a dominant tone.

Test 2. Input: Two sinusoids, approximately 3 kHz each.

Results: This display sounded similar to that of Test 1.

Test 3. Input: Low frequency noise, limited to frequencies below 350 Hz, plus two sinusoids as in Test 2.

Results: Three repetitions sounded alike, and similar to Test 1.

Test 4. Input: High frequency noise, bandlimited between 10 and 35,000 Hz, plus 2 sinusoids as in Test 2.

Results: Very little recognizable difference in three repetitions, all of which sounded similar to Test 1.

Considering the results of the above four tests, it was noted that there was no real change in the display for each test. If the system were performing as desired, this result would be expected. The addition of wideband noise should have no effect on the spectrum and hence on the display, other than to increase all levels. The two noise sources used were those available from the analog computer. According to the information available in Reference 8 these sources

produced Gaussian noise with spectrums of 0 to 350 Hz and 10 to 35,000 Hz for the low and high frequency noise generators respectively. Thus no effect would be expected from the low frequency generator since all of its frequency components would be far enough removed from the sinusoids as to contribute nothing to their spectral densities.

The spectrum of the high frequency generator is so wide that those components falling within the range of the sinusoids would be expected to have little effect.

Test 5. Input: Low frequency noise generator alone.

Results: On the first and third run of this experiment, a distinct high frequency component was heard in the display. On the second run, a lower frequency tone was heard, but in no case were the results similar to those produced during the first four tests.

Remarks: In order to verify that the displays produced in this test were different from those of the previous tests, Test 1 was repeated. It sounded quite similar to the results previously obtained, but did not resemble the results of Test 5.

Test 6. Input: High Frequency noise generator alone.

Results: With this input there was no similarity among the displays of six consecutive runs. In each case a different set of tones dominated the display.

Remarks: The display is sensitive only to relative magnitudes of 512 spectral components, and only the maximum spectral component and the seven surrounding components are chosen from those 512 samples. With such a wide spectrum (10 to 35,000 Hz) being sampled, and thus the relative maximum based on a very small sampling of the total spectral density, it is not at all surprising that the results should be different in each case.

Another run was made of Test 1 which confirmed that the system was still operating correctly.

Test 7. Input: Two sinusoids, 3000 and 3125 Hz.

Results: a. On the first run of this test it was difficult to determine if there was any significant difference between the result of this test and that of Test 1.

b. In order to determine whether there was a difference or not, the 3125 Hz generator was switched in and out on alternate runs. The resulting displays were significantly different in those cases where the 3125 Hz tone was present at the input in that the output appeared to drop in frequency relative to the center tone.

Remarks: This seems to indicate that the display could be used to differentiate between two distinct frequencies.

Test 8. Input: 3000 and 3125 Hz sinusoids plus high frequency noise generator.

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Results: The display sounded similar to that of Test 7

when both sinusoids were present at the input.

Test 9. Input: Same as Test 8 except that the 3125 Hz was switched

in and out on alternate runs as in 7.b.

Results: Similar to 7.b. A definite difference could be

noted when the 3125 Hz tone was present.

All of the above tests were repeated many times, always

with similar results.

3. Attempts to Simulate Sonar Signal

The third series of experiments on the system were essen-

tially the same as the previous group. In this case, however, the

noise was band limited to a passband between 2800 and 3200 Hz, in

an attempt to simulate bandlimited reverberation. A single signal

generator, set at approximately 3 kHz, was used as a signal source.

The results of a large number of runs showed that when only

noise was present, the resultant display was unpredictable. Various

combinations of tones were produced on each run. Sometimes as

many as all eight tones might occur, while on other runs only a

single tone or a few tones would be present in the display. The im-

portant fact, however, is that the display did vary. Only very rarely

did the same display occur twice in succession when only noise was

the input.

On the other hand, when the sinusoid alone was tested, the resultant display was never composed of more than three tones, and usually was only one or two. In addition, consecutive tests gave essentially the same results. Variations either occurred over a long period of time or when some change was made to the system, such as a change to the signal generator frequency.

With an input of both noise and the sinusoid simultaneously, the resulting display was similar to that of the display resulting from the sinusoid alone. Consecutive displays were similar in structure, and, although there were usually more tonal components present in these displays, generally one or two components dominated. It was also found that at the input the sinusoid could be masked so completely by the noise that it was not detectable, even when turned on and off. In spite of this masking, the system was still able to tell when the sinusoid was present. At this time, however, no measurement of minimum signal-to-noise ratio was made.

4. Attempts to Remove Mental Bias of Observer

The results obtained to this point of the experimentation were all performed with a degree of mental bias, as noted previously. The experimenter was performing each test with a certain amount of a priori knowledge of the display results. In other words, it was difficult to have a detached attitude concerning the output when the input was known. Some method of removing this bias was necessary in order to evaluate the system more fairly.

With this in mind, it was decided that a magnetic tape recorder whould be added to the system, so that a recording of many displays could be made for future use. This severely complicated the system because an appropriate recorder could not be located. Ideally, the recorder should be capable of almost instantaneous acceleration and deceleration so that display samples could be placed back-to-back for comparison and to form a continuous display. A basic problem with the entire system is the time required for processing. Each step in the program requires a discrete amount of time and as more steps are added, more time is required for their completion. This in turn causes a greater time lag between consecutive displays, increasing the difficulty in making comparisons. For example, an (unsuccessful) attempt was made to display the entire spectrum on the oscilloscope. This added 9 steps to the program and significantly increased the processing time.

To make a really useful recording it would be ideal to have a recorder which could be turned on and off by the display so that a continuous display would result. No such recorder was available. Another possible solution would be to store the DAC values in memory so that the previous display would continue while a new input sample was being processed. Such a procedure would not require a special recorder and would therefore prove invaluable. A method of implementing this idea was not discovered. One problem is that

the analog computer is taken out of the COMPUTE mode whenever the digital computer performs the FFT, thereby destroying the display. The portion of the digital program which controls the display is a sub-routine of the main program. The way the digital program is presently written, the computer must exit the subroutine and return to the main program in order to perform the A/D conversion.

A third solution which was considered was to make a continuous recording, cut out the empty portions of the magnetic tape, and then splice the remaining sections together to form a continuous display. It was decided that this method was impractical.

As a final compromise, an ordinary stereo tape recorder was used. One track was dedicated to the computer display and the other to annotation.

Several long series of recordings were made and played back. These recordings consisted of random combinations of signal alone, noise alone, and signal plus noise, in no particular order. As each recording was played back, an attempt would be made to determine what was represented by each sample. Unlike the case where the input conditions were known, the results of these experiments showed that the display was not always as clear in its presentation of information as had previously been thought. It was usually fairly clear when a signal without noise was the input, but there was often a great deal of difficulty in determining the difference between displays of a signal

plus noise input and an input of noise only. One clue to the difference was that in those cases where signal plus noise was present, the display tended to be constant over several repetitions, whereas the displays of inputs of noise alone usually varied in tonal content from one run to the next.

In attempt to clear up some of the problems in discerning the difference between various types of displays, a large number of samples of each input case (i.e., noise alone, signal alone, and noise plus signal) were taken. Examination of the line printer output showed that with the DAC settings ranging between 0.0000 and 1.0000, an average of two DAC lines per run were set to less than 0.3 when only noise was present at the input. In contrast to this, an average of four lines were set to less than 0.3 when signal plus noise was the input, and an average of six were below 0.3 when signal alone was the input. This suggested adding a step to the program to suppress or set to zero all DAC line outputs of less than 0.3. Such a step was added to the program and resulted in clearer sounding tones, and in some cases made it easier to tell when a signal was added to noise.

5. Quantitative Experiments

The final series of experiments was concerned with attempts to gather data of a more quantitative nature in order to better judge the capabilities of the system. For this phase of the experimentation, 200 separate runs were made. For each run, after signal and noise levels were set, the input signal was first displayed in order to determine if

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the signal could be heard through the noise when both were present.

The peak-to-peak voltage level of the sinusoidal signal was raised in 0.5 volt steps from 1.0 V PP to 8.0 V PP, and in 1.0 volt steps from 8.0 V PP upwards. These values were measured on the installed oscilloscope. The noise generator input to the system was passed through a Krohn-Hite model 3750 bandpass filter. This filter allowed control of the upper and lower rolloff slopes of the bandpass characteristic in 6 dB/octave steps from 6 to 24 dB/octave. The gain of the filter was set at 20 dB for all experiments. This caused the noise level to be clipped at 10 V PP. The signal generator was set at 3 kHz for all experiments. All other parameters and system controls were set as previously described.

6. System Performance Results

The results of these experiments may be summarized in two parts. The first part concerns the system performance in discriminating between the four input conditions - no input signal or noise, noise alone, signal alone, and noise and signal.

a. No signal or noise

This experiment was performed twice, and on both occasions, as had occurred in previous tests, the system displayed only the lowest tone of the encoded display. This was an indication that the maximum spectral component occurred at DC, i.e., there were no components below the maximum.

b. Noise only

The results of these tests were similar to those previously noted for this input case. Any number of components might appear in the display. In some cases only a few would be present, but there was no similarity among runs.

c. Signal only

In 15 runs of this experiment, the system always met the detection criteria, which had been defined as a display containing three or fewer tones. In fact, when only a signal was present at the input, the system displayed only a single tone in 12 of the 15 cases. In the other three cases, two tones were presented, and in each of these displays, the second tone was adjacent to the center tone and had a magnitude which was less than 45% of that of the center tone. The detection criteria was met with as little as 1 V PP signal input, the minimum level used.

d. Signal plus noise

(1) Reliability of Detection. Again the detection criteria was defined as three or fewer tones displayed to indicate a signal present in the noise. Although the system gave some correct responses with as little as 2.5 V PP (SNR = 0.25), the responses were not considered reliable until a signal level of 13 V PP (SNR = 1.3) was reached. Reliable response was defined for these purposes as being a minimum of three consecutive correct responses by the system.

At 13 V PP there were 23 runs made and of these, the system res-
ponded correctly 19 times (83%). All of these tests were made with
the noise filter set to pass 2800 to 3200 Hz with a roll-off character-
istic of 24 dB/octave on either end of the passband. For similar con-
ditions with a 14 V PP input, there were 24 attempts. Correct
responses occurred 21 times (87.5%).

(2) Effects of Filter Roll-off Characteristics. With the
input signal set at 14 V PP, the effect of the filter roll-off character-
istics was next investigated. Since the probability distribution of the
amplitude of sonar reverberation is a Rayleigh function [Urick, p 226],
a more gradual roll-off characteristic would seem to be a better
approximation. Accordingly, the roll-off was reduced in 6 dB steps
as provided by the filter. With the passband set at 2.8 to 3.2 kHz,
the following results were obtained with a signal plus noise input, and
the signal input set at 14 V PP:

Roll-off	Number of Runs	Number of Successes	Percent Success
24 dB/octave	24	21	87.5
18 dB/octave	6	5	83.3
12 dB/octave	11	9	81.8
6 dB/octave	13	8	61.5

If the 6 dB/octave case is ignored, it appears that for this small
population sampling the results are essentially independent of the
shape of the passband. On the other hand, it might also be said that
there is a general downward trend in the results as the bandpass

characteristics become less steep. It is difficult to base any firm conclusion on such a small sampling.

(3) Effect of an Increased Passband. As a final test, the passband was increased from 400 Hz (2.8 to 3.2 kHz) to 800 Hz (2.6 to 3.4 kHz), with the roll-off characteristic set at 24 dB/octave. The results are tabulated below.

Passband	No. Runs	No. Success	% Success
2.8 - 3.2	24	21	87.5
2.6 - 3.4	10	9	90.0

Again it appears that there is essentially no difference. However, as noted above, the small population does not allow a definite conclusion to be drawn.

7. Man-Machine Interface Effectiveness

a. Method

The above results indicate, in general, that the system is able to make a decision about the presence or absence of a sinusoid in noise with a certain degree of reliability. The second part of this series of experiments was to determine whether the system would be able to pass that decision on to the human observer through the inter-face of the auditory display. To learn this, during the 200 runs of the experiment which supplied the data for the above results, the system was also observed at its audio input in each case to determine if the signal could be detected in the noise, and at its audio output to determine what information could be learned from the display. For

each display, an attempt was made to count the number of signals present in the display, and where it was possible, the number was recorded. When not possible, some other descriptive phrase was used so that for each of the 200 runs there would be some sort of data which could be compared with the line printer output. Since the line printer is physically located on the opposite side of the computer laboratory from the analog computer console, it was not possible to make an immediate comparison. Thus, in this case, the mental bias noted before was possibly reduced.

b. Results

The following observations were made when the written comments were compared with the line printer output:

(1) When only one tone was present, it was obvious, and was invariably correctly identified as being a single tone. This result is trivial.

(2) In those cases where two tones were displayed, they were recognized as two separate tones only in certain instances. Designating the center tone (normalized to the maximum value as noted before) as the first or reference tone, the second tone would be recognized if:

      (a) it was not adjacent to the first or center tone.

      (b) it was adjacent to, and higher in frequency than the center tone.

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(c) it was adjacent to, and lower in frequency than the center tone, and had an amplitude greater than about 40% but less than about 50% that of the center tone. If the second tone had a magni- greater than about 50% of the first tone, another tone, possibly a beat note, was often heard. If the tone amplitude was between 30 and 40% of the center tone, it was not heard. As noted above, tones with less than 30% the amplitude of the cen- ter tone were suppressed by the system.

(3) Similar results occurred when three tones were present.

(4) In no case was a correct identification made as to the number of tones present when the display was composed of more than four tones. In such cases the display was described as having "many" or "several" tones. In some cases it was possible to determine that tones lower or higher than the center tone dominated the display.

(5) In many cases, when the display comprised more than three tones, masking of adjacent tones would result in fewer tones being heard.

(6) In those cases where the input was changed by deleting the noise or the signal, it was necessary to observe at least three repe- titions before it was possible to determine with any certainty what

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result the computer was displaying. When signal and noise or signal only was the input, the resulting display, although varying somewhat in composition from one repetition to the next, generally retained the same number of tones in its composition. In those cases, however, where noise was the only input, the resulting display varied widely from repetition to repetition in number and amplitude of tones present.

(7) Considering only those cases where the signal input was greater than 13 V PP, there were 88 runs in which both signal and noise were present. By the previously established criteria that three or fewer tones in the display constituted a correct response by the system, there were 71 correct responses (80.6%). Of these 55 (77.5%) were correctly identified by the observer as to number of tones present. Of the 16 incorrect responses, 12 were too low (i.e., recognizing one tone when two were present or two when there were three present) and four were too high (hearing three when two were present, etc.). There were no errors when the display was composed of only one tone. The errors on the low side were most likely due to masking. This was also confirmed by those cases where more than three tones were present but fewer tones were heard. Those cases where more tones were heard than were actually present in the display were most likely the consequence of hearing beat notes.

(8) There was no case in any of the 200 runs where the input signal could be heard through the noise. On several tests when

the signal was turned on and off, it was possible to tell that something was happening, but it was not possible to tell that a signal was being added to or deleted from the noise input.

IV. CONCLUSIONS

A. SUMMARY OF RESULTS AND CONCLUSIONS

The experiments conducted during the course of this research indicated that the proposed acoustic display system has some merit. Problem areas, previously unanticipated were uncovered, and areas for future research were discovered.

Two aspects of such a display were concurrently examined. These were (a) the signal processing system which attempted to locate a desired signal in noise, and, (b) the display of the processed information in the form of a man-machine acoustic interface.

The digital signal processing program performed the desired function passably, at least to the extent that it was capable of determining the difference between signal and no signal conditions when no noise was present. The addition of noise reduced the system effectiveness. The major limitations on the system were imposed by the characteristics of the XDS 9300/CI 5000 computer system and the program which was used. The maximum sampling rate (10 kHz) and the memory size which limited the number of samples to 1024 resulted in a long processing time (about 0.1 sec) and a resulting frequency spectrum resolution of about 10 Hz. This is somewhat coarser than might be desirable. In the case of a 3 kHz sonar, a 10 Hz shift in frequency would indicate a target speed on the order of 5 knots. The detection of a slower target would be preferable.

The minimum signal-to-noise ratio for a reliable determination of the presence of a signal in noise was found to be on the order of about 1.3. This is a voltage ratio and converts to about 2.3 dB. The power of the signal must therefore be almost twice that of the noise. In order to be considered effective a considerably lower signal-to-noise ratio is desirable. Camp [2] provides excellent information on various types of sonar signal processor systems, including spectral analysis methods, in which he discusses SNR figures on the order of -15 dB or smaller. However, he is dealing with more complex systems which include many more factors than a simple voltage ratio of a threshold of hearing. Although in this system the signal could not be heard through the noise, the threshold of hearing was not very much greater. The subject of hearing a tone masked by noise and threshold effects is discussed in any text on applied psychology or human engineering, including refs 4, 6, 9, 12. The fact that the sound could not be heard, moreover, was a subjective evaluation, and may not have been true for an observer with better hearing.

The research on the display formed the greater part of the effort. Prior to the beginning of the research it was not known how effective this particular display would be or even how it would sound. Work has been done on multi-dimensional audio displays by several researchers [6, 11, 13], but no reference was found to indicate that a display such as that proposed by this thesis had been attempted.

Although the display, as implemented, can be made to work, its usefulness in the present form is questionable. The display has been found to be basically binary, answering a yes-no question. Either the signal is present or it is not. If this were the only information desired from the display, far less elaborate methods are available for producing an auditory yes-no display. No more than a buzzer or a single oscillator would be necessary to indicate the presence of a signal (or, in the case of a sonar system, a possible contact). The intention of the research was to discover a system of presenting more than one bit of information. Evidently, using a number of tones may not facilitate this. Part of the problem is due to the fact that the processing method used to generate the display only presented one bit of information (signal present) and any other information which might have been displayed was not generated. The other part of the problem is in the display itself. The combinations of tones were often confusing, and they required a great deal of concentration to interpret not only tonal combinations, but also amplitude variations.

When the system had no input, the lowest frequency tone was presented, indicating that the spectrum contained only a DC component. When a signal without noise was present at the input, the center tone was displayed (or on some occasions several centered tones). This information would be useless without a reference. For example, if the observer were to hear a tone upon initially activating the system, he

would not know what the significance of that tone was until the opposite case was presented to him. As noted before, this is a yes-no situation wherein a change in the display indicates only that something has occurred.

When noise is added to the input, more tones are presented. This is a third case, again requiring a comparison. The display becomes one of degree. If there is no input, a single low frequency tone is displayed. If a signal is present at the input, one higher frequency tone results. The addition of noise adds more tones, and the number of tones added has some correlation with the amount of noise present. In this respect the system becomes multidimensional, indicating presence or absence of three quantities (nothing, signal, noise), and a ratio of two of those quantities (signal, noise). In some applications this might prove useful.

However, for purposes of attempting to encode the information presented by the processing system, the resulting display tended to cause confusion. Several repetitions of the display were necessary before a decision could be made. This, of course, is a function of the signal processing system as well as the display. The reliability of those decisions, once made, was questionable. One goal of the research was to produce a display which did not require constant attention. Because comparisons of consecutive displays were required, it was necessary to listen very closely to each display. A change in quality or number of tones was not a reliable indication that some change had occurred at the input.

Additionally, constant attention to this display was tiresome, even after the encoding had been tuned to a musical scale.

These factors tend to indicate that the display would not be desirable in situations where an operator was devoting less than his entire attention to it. It was noted that headphones were required in order to properly hear the display and avoid external interference. The system would be undesirable on the bridge or in C.I.C. of a ship as it would either add to the noise level or require one more "body" to be present.

In order to be effective, it is obvious that a display must present information in an identical manner to all observers. If this is not true, and if interpretation is possible or even necessary, the effectiveness of the display is decreased. Such was the case with this display. Throughout the experiments it was noted that it was possible to hear tones not actually present. This amounts to the auditory equivalent of an optical illusion. It may be possible to capitalize on this illusion through a different encoding scheme, but in the present form it increases confusion in understanding what is being displayed.

In summary, it is felt that the combination of the digital processing system and the display as designed and implemented fell short of performing the desired function. This statement holds several challenges. First of all, efforts should be made to improve the processing system so that it is better able to locate a signal in noise. One solution to this problem is to make use of a different computing system. The XDS

9300/CI 5000 system was used because it was available, and because it was felt that it would afford a degree of flexibility in implementing the display system. But the slowness of the digital computer, the small memory size, and the instabilities of the analog computer all contributed to the design problems. A far better solution would be to use a dedicated minicomputer or a spectrum analyzer. Suitable instruments are available commercially.

The instabilities of the analog computer, the number of amplifiers available, and difficulties inherent in attempting to interface with the computer output could all have been avoided if the oscillators were built using ordinary electronic construction methods. This was not possible, however, since in the manner in which the system was implemented, the XDS 9300 could not have been used to directly control external oscillators.

Finally, it is felt that the display system has merit. Although it it appears that in this particular application it merely caused confusion, if the output of the digital portion had been more clearly defined, the display would have been more meaningful. For example, if the system had been able to indicate the difference between a signal in noise and noise alone as clearly as it was able to show the difference between signal and no signal conditions in the absence of noise, it would have proven more useful. Or, if its task were not one of detection, but rather one of indicating relative amounts of something such as signal and noise, it could have performed much more acceptably.

Thus the display itself is usable and has value, although possibly not in this application.

B. POSSIBLE FUTURE RESEARCH

If a system to signal the presence of a sonar target is to be developed, and if an audio display is to be used, some different form of display is probably necessary. One possible display might be some sort of indication similar to a Morse code character. This would require a complete redefining of the display as well as an entirely different interface between processor and display. Such a display might take the form of a series of "dits" to indicate the presence of a target in a range band or on a given bearing. A series of one to four "dits" might indicate the relative likelihood of a target's presence. Thus, one "dit" would indicate no target present, with an increasing number of "dits" indicating the relative probability, and four indicating a definite contact. Such a system indicates at least two bits of information (presence or absence of a possible contact and reliability of the contact). In addition the number of "dits" displayed might have a feeling of urgency associated with it, one "dit" meaning no danger, four indicating a great degree of danger. A display of this form might also be usable in an area other than the sonar spaces without causing interference to other activities.

APPENDIX A

ANALOG COMPUTER COMPONENTS AND SETTINGS USED

A. Components used. Refer to Figure 6.

OSCILLATORS	AMPLIFIER				POTENTIOMETER	
	A1	A2	A3	2	LIMIT	DAMP
1	A001	A003	A000	P002	P401	P000
2	A005	A007	A006	P006	P403	P005
3	A013	A015	A014	P014	P405	P013
4	A021	A023	A022	P022	P407	P021
5	A025	A027	A026	P026	P411	P027
6	A033	A031	A034	P034	P413	P031
7	A035	A041	A042	P042	P415	P043
8	A043	A047	A044	P044	P417	P045

Input Summing Amplifier: A011
Output Summing Amplifier: A076

By taking the output from the READ BUSS (Ref. 8) it is possible to examine the output of any amplifier by entering the amplifier number at the analog console keyboard. For example, optimum settings for the limit pots may be determined by entering the amplifier number at the console. The use of the keyboard also allows examination of system input and outputs.

B. Initial potentiometer settings (100 Hz frequency spacing)

OSC		2	DAMP		LIMIT	
1	P002	0.3551	P000	0.0246	P401	0.8351
2	P006	0.6316	P005	0.0211	P403	0.8361
3	P014	0.9868	P013	0.1223	P405	0.8404
4	P022	0.1420	P021	0.0274	P407	0.8364
5	P026	0.1934	P027	0.0524	P411	0.8367
6	P034	0.2527	P031	0.0692	P413	0.8383
7	P042	0.3199	P043	0.1576	P415	0.8365
8	P044	0.3948	P045	0.0719	P417	0.8346

[2]settings are calculated using the equation $^2 = (2\ f)^2$.

Settings above are for 100 Hz spacing from 300 to 1000 Hz.

Damping potentiometer settings are experimentally determined to provide a stable sinusoidal output.

Limit potentiometers are set to provide a 40 volt peak-to-peak output from each oscillator.

C. 2 settings for musical notes from G to G^1. (See Appendix B)

NOTE	OSC	POT	SETTING
G	1	P002	0.6065
A	2	P006	0.7644
B	3	P014	0.9630
C^1	4	P022	0.1080
D^1	5	P026	0.1362
E^1	6	P034	0.1716
F^1	7	P042	0.1926
G^1	8	P044	0.2426

Other potentiometer settings are the same as in B above.

D. 2 settings for musical notes from C to C^2 to spread frequency encoding.

NOTE	OSC	POT	SETTING
C	1	P002	0.2702
E	2	P006	0.4279
G	3	P014	0.6066
B	4	P022	0.9629
D^1	5	P026	0.1362
F^1	6	P034	0.1926
A^1	7	P042	0.3057
C^2	8	P044	0.4324

Note that whenever 2 settings are changed, it may be necessary to also change the integrator capacitors.

E. Final limit potentiometer settings.

OSC	POT	SETTING
1	P401	0.7076
2	P403	0.7435
3	P405	0.7570
4	P407	0.7990
5	P411	0.8047
6	P413	0.8564
7	P415	0.9363
8	P417	0.9546

These values provide an equal output level (subjective) for all oscillators.

APPENDIX B

MUSICAL SCALE AND ASSOCIATED ANALOG COMPUTER

POTENTIOMETER SETTINGS

POTSET lists the four-place decimal potentiometer setting.   The exponents are set by proper choice of the integrator amplifier capacitors.

NOTE	f(Hz)	POTSET	EXPONENT
C_1	130.81	0.6755	6
D_1	146.83	0.8511	6
E_1	164.81	0.1072	7
F_1	174.61	0.1204	7
G_1	196.00	0.1517	7
A_1	220.00	0.1911	7
B_1	246.94	0.2407	7
C	261.63	0.2702	7
D	293.66	0.3404	7
E	329.63	0.4279	7
F	349.23	0.4815	7
G	392.00	0.6066	7
A	444.00	0.7643	7
B	493.88	0.9629	7
C^1	523.25	0.1081	8
D^1	587.33	0.1362	8
E^1	659.26	0.1716	8
F^1	698.46	0.1926	8
G^1	783.99	0.2427	8
A^1	880.00	0.3057	8
B^1	987.77	0.3852	8
C^2	1046.50	0.4324	8
D^2	1174.70	0.5448	8
E^2	1318.50	0.6863	8
F^2	1396.90	0.7704	8
G^2	1568.00	0.9706	8

APPENDIX C

INSTRUCTIONS FOR USE OF THE DISPLAY PROGRAM WITH THE

XDS 9300/CI 5000 COMBINATION

1. Install prepared patchboards on CI 5000.

2. Install tape on tape recorder.

3. Compile digital program.

4. Connect inputs and outputs to patchbay. Input is to

    amplifier A011; output is from A076.

5. Put CI 5000 in DIGITAL control.

6. When XDS 9300 indicates it is ready (entry light at TTY),

    type:

NREC	=	1	(number of records)
NSAMP	=	1024	(number of samples)
NCHAN	=	1	(number of channels)
NPOT	=	16	(Number of potentiometers to be set)
NDEL	=	1	
ITAPE	=	1	(tape channel as set on tape machine)

    * C/R

7. Set manual pots. When digital computer has completed

    setting syncro pots it will indicate manual pots are to be

    set. When done, type *C/R to continue digital program.

8.   When digital computer types OPTION = (I1), enter the
      number corresponding to the desired option. Option 2
      is the normal operating mode.

      Option 1 returns control to the point in the program where
      changes may be made to the TTY inputs as in step 7 above.
      For example, different values of pot settings may be made.
      It is not necessary to re-enter all values. Other options
      are available but are not used for the system operation.
      For their use consult the program.

9.   When option 2 has been selected. The system is ready
      to operate. Set the following controls at the analog
      computer console:

      a.   Preset Counter = 09000000.

      b.   DF00 = 0.1 msec.

      c.   DF01 = 0.1 sec.

      d.   DF02 = 0.1 sec.

10.   To operate the program, momentarily depress toggle
      switch DS0. To stop the process, depress DS1. This
      returns the program to the point where DS0 will initiate
      another run. If DS2 is turned on and then DS1 is momen-
      tarily depressed, the program will return control to the
      TTY and will request a new OPTION entry.

NOTE: The specific switch settings and amplifiers were those

used in the program as implimented for this research.

They depend upon the wiring of the analog patchboard.

```fortran
      DIMENSION IBUF(1200),LOCB(-1:1),BADREC(100)
      INTEGER RECNUM,BADREC
      NAMELIST NREC, NSAMP, NCHAN, ITAPE,NDEL,NPOT
1     INPUT(101)
      LOCB(-1)=LOCF(IBUF(1))
      LOCB(1)=LOCF(IBUF(1025))
      NWORDS=NSAMP*NCHAN
      DO 510 I=1,NPOT
      READ(5,500)(IPOT,POTV)
500   FORMAT(A4,F1C.4)
      CALL SETPOT(IPOT,POTV)
510   CONTINUE
      OUTPUT(101)'SET MANUAL POTS'
      INPUT(101)
      GO TO 15
2     NB=1
      RECNUM=0
      NEWBUF=LOCB(1)
      CALL WRITECLOCK(O)
      CALL ADSTART(NCHAN,LOCB(-1),NEWBUF,NSAMP,RECNUM,11S)
      CALL MTRDY(ITAPE,LOCB(-1),LOCB(1),NWORDS,IND)
4     IF IND.EQ.1, GO TO 4
      GO TO(4,3,91,93)IND
3     IF(TEST(1).GT.0)GO TO 3
      NBAD=0
      CALL STARTCLOCK
      CALL ENABLE
5     CONTINUE
10    GO TO 5
11    NB=-NB
      NEWBUF=LOCB(NB)
      IF(TEST(1).GT.O.OR.RECNUM.GE.NREC)CALL DISABLE
      GO TO(90,12,91,92)IND
12    CONTINUE
      CALL MTOUT(NB)
```

```
      IF(TEST(1).LT.0.AND.RECNUM.LT.NREC)GO TO 5
      CALL STOPCLOCK
      CALL ADSTOP
      CALL PROCESS (IBUF,NSAMP,2S)
      OUTPUT(101)RECNUM
      IF NBAD.NE.0,WRITE[6,106] [BADREC[I],I>1,NBAD]
106   FORMAT[$ BAD RECORDS ARE$,[10I6]]
15    OUTPUT(101)'OPTION=(I1)'
      READ(101,100)NOPT
100   FORMAT(I1)
      GO TO(1,2,30,40,50,60,70)NOPT
30    ENDFILE(ITAPE)
      OUTPUT(101)'EOF'
      GO TO 15
40    REWIND(ITAPE)
      GO TO 15
50    OUTPUT(101)'SKIPFILES=(I4)'
      READ(101,101)NF
101   FORMAT(I4)
      DO 55 I=1,NF
51    CALL BUFFERIN(ITAPE,1,IBUF(1),1,IND)
52    IF(IND.LT.2)GO TO 52
      IF(IND.NE.3)GO TO 51
55    CONTINUE
      OUTPUT(101)NF
      GO TO 15
60    OUTPUT(101)'NUMWORDS TO LIST=(I4)'
      READ(101,101)NW
      WRITE(101,105)NW,NCHAN
105   FORMAT(' WRITE ',I4,' WORDS, ',I2,' AT A TIME')
      IND=1
      CALL BUFFERIN(ITAPE,1,IBUF(1),NWORDS,IND)
66    IF(IND.EQ.1)GO TO 66
62    GO TO(62,63,64,65)IND
63    WRITE(6,102)
```

```
102   FORMAT(1H1)
      DO 631 I=1,NW,NCHAN
      WRITE(6,104)(IBUF(J),J=I,I+NCHAN-1)
104   FORMAT(12010)
631   CONTINUE
      GO TO 15
64    OUTPUT(101)'EOF READ'
      GO TO 15
65    OUTPUT(101)'READ ERR'
      GO TO 63
70    OUTPUT(101)'START ANALOG RECORDER'
      OUTPUT(101)'TYPE * C/R TO CONTINUE'
      INPUT(101)
77    IND=1
      CALL BUFFERIN(ITAPE,1,IBUF,NWORDS,IND)
76    IF(IND.EQ.1)GO TO 76
71    GO TO(71,72,64,74)IND
72    DO 73 I=1,NWORDS
73    IBUF(I)=IBUF(I)/2**10
      DO 75 I=1,NWORDS,NCHAN
      DO 750 J=1,NCHAN
750   CALL DAC(J,IBUF(I+J-1))
      N=NDEL
      CALL DELAY
75    CONTINUE
      IF(ISENSE SWITCH 1)77,15
74    OUTPUT(101)'READ ERROR'
      GO TO 72
90    CALL DISABLE
      CALL ADSTOP
      OUTPUT(101)'RATE ERR',RECNUM
      GO TO 15
92    NBAD=NBAD+1
      BADREC(NBAD)=RECNUM-1
      GO TO 12
```

70

```
91      CALL DISABLE
        CALL ADSTOP
        OUTPUT(101) ' MT NOT READY '
        GO TO 15
93      CALL DISABLE
        CALL ADSTOP
        OUTPUT(101) ' MT ERROR ON SPACING FROM LOAD POINT'
        GO TO 15
        END
```

```fortran
      SUBROUTINE PROCESS (ISAMPLE, NUM, LOC)
      DIMENSION DATA(2,1024), ISAMPLE(1024), SPEC(512), V(12)
      DO 100 I=1,NUM
      DATA (1,I) = ISAMPLE(I)
  100 DATA (2,I) =0
      CALL FOUR2(DATA,NUM,1,-1)
      SPECMAX = 0
      DO 200 I=1,NUM/2
      SPEC(I)=SQRT(DATA(1,I)**2+DATA(2,I)**2)
      IF(SPEC(I).LE.SPECMAX) GO TO 200
      SPECMAX=SPEC(I)
      K=I-4
      IF(K.LT.0)K=0
  200 CONTINUE
      DO 400 L=1,12
      KL=K+L
      V(L)==SPEC(KL)/SPECMAX
  400 IF(ABS(V(L)).LT.0.300)V(L)=0
      WRITE(6,500)(V(L),L=1,8)
  500 FORMAT(8F10.4)
      CALL DAC(1,V(1),2,V(2),3,V(3),4,V(4),5,V(5),6,V(6),
     1 7,V(7),8,V(8))
      CALL RESET (100)
      CALL COMPUTE
  300 IF(TEST(2).GT.0) GO TO 300
      CALL RESET(100)
      IF (TEST(3).GT.0) RETURN
      RETURN LOC
      END
```

```
$ADSTART PZE
         BRM  9SETUPN
         PZE  6

NCH      PZE
BUF      PZE
NEWBUF   PZE
NSAM     PZE
RECNUM   PZE
NEXLOC   PZE
         LDA  ENDBRM
         XMA  040
         STA  SV040
         LDA  INTBRM
         XMA  052
         STA  SV052
         LDA  *NCH
         STA  NCH
         STA  INCR
         ADD  =COMM
         STA  COMLOC
         LDA  NCH
         LLSA 15
         ADD  =COMM
         STA  *COMLOC
         STA  CONTR
         LDA  *BUF
         STA  COMM
         STA  CLKPTR
         LDA  *NSAM
         SUB  =1
         STA  COUNT
         STA  NEWCT
         EOM  034001
         POT  CONTR
         BRR  ADSTART
```

```
INTBRM  BRM  ADFAST
ENDBRM  BRM  ENDAD
SV040   PZE
ENDAD   PZE
        DIR
        EOM  034001
        POT  CONTR
        EIR
        BRC  *ENDAD
$ADSTOP PZE
        LDA  SV040
        STA  040
        LDA  SV052
        STA  052
        STZ  *COMLOC
        MPO  ADSTOP
        BRR  ADSTOP
COMLOC  PZE
$ADFAST PZE
        DIR
        STD  SVAB
        LDA  INCR
        ADM  COMM
        LDA  R\CLOCK
        ETR  *01777
        ADM  *CLKPTR
        LDA  COMM
        STA  CLKPTR
        SKR  COUNT
        BRU  EXIT
        LDA  *NEWBUF
        STA  COMM
        STA  CLKPTR
        LDA  NEWCT
        STA  COUNT
```

```
        MPO    *RECNUM
        LDP    SVAB
        EIR
        BRC    *NEXLOC
        LDP    SVAB
        EIR
EXIT    BRC    *ADFAST
COMM    RES    33
CONTR   PZE
NEWCT   PZE
SVAB    RES    2
INCR    PZE
COUNT   PZE
CLKPTR  PZE
SVO52   PZE
        END
```

```
A       EQU     5
B       EQU     4
X1      EQU     1
$MTRDY  PZE
        BRM     9SETUPN
        PZE     5
UNIT    PZE
BUF1    PZE
BUF2    PZE
NW      PZE
IR      PZE
I       EQU     IR
        LDB     *NW
        COPY    (0,A),(B,X1)
        LLSD    14
        MRG     ALCO
        STA     ALCN1
        STA     ALCN2
        COPY    (0,A)
        LDB     *BUF1
        LLSD    10
        LLSA    5
        ADM     ALCN1
        COPY    (X1,A)
        LLSD    14
        STA     8UTCW1
        COPY    (0,A)
        LDB     *BUF2
        LLSD    10
        LLSA    5
        ADM     ALCN2
        COPY    (X1,A)
        LLSD    14
        STA     8UTCW2
        LDA     =1
```

```
        STA    *IR
        LDB    *UNIT
        LDA    TRTO
        COPY   7,(B,A)
        STA    TRTN
        LDA    FPTO
        COPY   7,(B,A)
        STA    FPTN
        LDA    BTTO
        COPY   7,(B,A)
        STA    BTTN
        LDA    WTBO
        COPY   7,(B,A)
        STA    WTBN
        LDA    EFTO
        COPY   7,(B,A)
        STA    EFTN
        BRM    RDY
        BRU    RDYERR
        EXU    FPTN
        BRU    RDYERR
        EXU    BTTN
        BRM    BTSP
        MPØ    *IR
        BRR    MTRDY

RDYERR  MPT    *IR
        BRR    MTRDY

*
* RDY   PZE    TRTN
        EXU    0
        CAT    RDY
        BRR    RDY
        MPØ    RDY
        BRR    RDY
```

```
*
BTSP    PZE     EORBRM
        LDA     O11
        XMA     SV11
        STA     EFTN
        EXU     ALCO
        EXU     BTGAP
        POT     BTSP
        MPO     BTSP
        BRR     BTSP
IOC     FORM    10,14
BTGAP   IOC     417,0
*
$MTOUT  PZE     9SETUPN
        BRM     1
        PZE
NB      PZE
        LDA     *1
        STA     *I
        BRM     RDY
        BRU     ERR
        LDA     EORBRM
        XMA     O11
        STA     SV11
        SKN     *NB
        BRU     OUT2
        EXU     WTBN
        EXU     ALCN1
        POT     OUTCW1
        BRR     MTOUT
OUT2    EXU     WTBN
        EXU     ALCN2
        POT     OUTCW2
        BRR     MTOUT
ERR     MPT     *I
```

```
        BRR    MTOUT
*
EOR     PZE    RDY
        BRM    $+2
        BRU    0
        CET    *I
        MPT    *I
        MPO    SV11
        XMA    O11
        STA    SV11
        XMA    *EOR
        BRC    EOR
EORBRM  BRM
SV11    PZE
*
TRTO    TRT    0,0
TRTN    PZE
FPTO    FPT    0,0
FPTN    PZE
BTTO    BTT    0,0
BTTN    PZE
WTBO    WTB    *0,0,4
WTBN    PZE
EFTO    EFT    *0,0,4
EFTN    PZE
ALCO    EOM    016000
ALCN1   PZE
ALCN2   PZE
OUTCW1  PZE
OUTCW2  PZE
        END
```

|	
PO45	0.0719
PO43	0.1576
PO31	0.0692
PO27	0.0524
PO21	0.0279
PO13	0.1223
PO00	0.0246
PO05	0.0211
PO02	.2702
PO44	.4324
PO34	.1926
PO26	.1362
PO14	.6066
PO06	.4279
PO42	.3057
PO22	.9629

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